HOW THE SCHOBER REVERBATAPE* UNIT WORKS

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There is almost no musical instrument or ensemble which does not sound better in a hall having some reverberation than in an acoustically dead room. This is especially true of an organ — pipe or electronic — since the organ is the most mechanical of all instruments and there is a great deal of "machinery" between the player and his audience (even if the organist himself is his only audience).

Reverberation is the "sound of a large auditorium" which is brought about by the bounces of the original sound against the walls of large, live halls, with consequent audible repetitions of each sound at each bounce. The total effect caused by the reverberatory bounces is an apparent slow decay of the sound after each note has ended, rather than the abrupt, cold harshness obtained in a small, padded room where everything is heard with razor-sharp precision and immediacy. Electronic organs, with their scientific precision of sound, are too precise and impersonal to begin with. The defocusing and softening effects of reverberation are really a necessity to give both player and audience the emotional satisfaction and sense of bigness so attractive in a theatre or church organ.

Though not easy to achieve in practice, the requirements for an artificial reverberation system are simply stated. It is only necessary to pass the sound electronically through a device which will repeat each sound at short intervals, each repetition somewhat lower in volume than the last, until it finally dies away. This is done by the spring-type units which are available today. However, if real satisfaction is wanted, the effect must sound like a natural one; that is, the music should have the same aural and emotional effects as in a real theatre or church, which means in effect that it must fool the hearer, rather than being obvious as just another sound effect. Realism in a reverberator is the ultimate goal, and un-

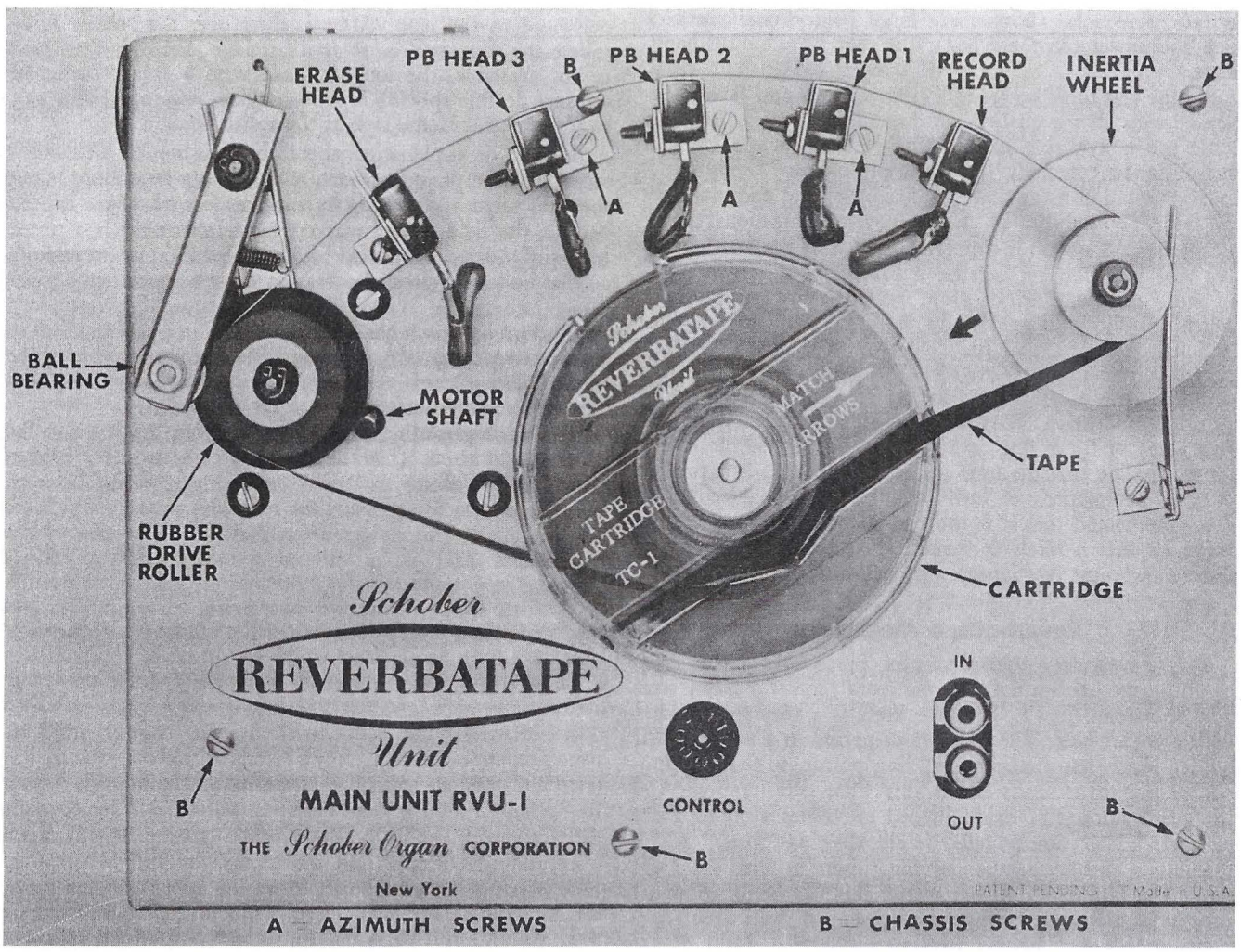

Fig. 1 — Main section of the Schober REVERBATAPE Unit contains tape mechanism and electronics. The plastic cover, which protects the mechanism and tape, has been removed.

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fortunately spring devices, which offer the maximum simplicity in design and the lowest cost, have inherent limitations that make realism impossible. They have sharp resonances, narrow bandwidth-handling abilities (no bass or high treble), and they sound, if anything, like very small rooms with more reverberation than a small room is likely to have, because the period between repetitions is too short. They also cannot be controlled to simulate realistically rooms of various sizes or with walls of varying hardness.

The best method of artificial reverberation is to use a continuous recording device, such as a tape recorder. The sound is recorded on the tape and is played back several times as a given section of tape passes several playback heads. This system has been in use for years in recording and broadcast studios. It has the advantages of complete controllability, in both design and use. It can have full frequency range, low distortion, any timing and reverberation-control characteristics. A good tape reverberator can produce acoustic effects indistinguishable from the real thing. Many phonograph records have been made in small studios, the acoustics enhanced by tape reverberators, without any knowledge on the part of listeners that the job was not done in a large concert hall!

Tape reverberators previously available, however, have been highly expensive — costing $1,000 to $3,000. They are customarily built with the most expensive components, are completely variable both mechanically and electronically, and employ the type of construction one would expect for hard professional use in a very large variety of different applications. An important factor in their pricing has been that so few were sold (how many large professional users of such equipment can there be?) that they were almost handmade.

For the first time, however, a relatively low-cost tape reverberator with all the quality and durability necessary for daily home use is available in the Schober Reverbatape Unit. It was designed especially for electronic organs(though it is equally usable in many other applications) and does not have the wide adjustability of the expensive professional units. Its mechanical system is simpler than that of the least expensive tape recorder; yet because it has only one job to do — no stop, reverse, fast, or slow, record, erase, or playback switching — it is as reliable as the good conventional professional recorders and has low wow and flutter content. Electronic circuitry is relatively simple, with no large power supply or power amplifier, and cost is further reduced by quantity production. It is still considerably more expensive than many of the spring reverberators, but its audible results are so realistic that the user quickly forgets that it is present — until he happens to turn it off! Professional musicians have commented that it is the one advance in the electronic organ art that is likely to make the electronic organ acceptable to the most reactionary pipe organist.

Reverbatape Mechanics

The Reverbatape Unit consists principally of a tape-handling mechanism built on the front face of a heavy steel box, shown in Fig. 1. The tape itself is a very special kind with a Mylar base. The oxide is dispersed in a new kind of binder which has enormous holding power, so that continuous passing over heads causes very little deterioration. And the small amount of oxide rubbed off comes away as a dry powder, which simply falls, rather than as "gunk" which remains in the head gaps and fouls them as with the usual tape. The tape itself is guaranteed for 500 hours of actual use; in laboratory tests, tapes have been run well over 2,000 hours before any audible effect has appeared. Most of the 22-foot endless loop is contained in a plastic cartridge of special design.

The tape is driven by a small 4-pole motor within the box, whose steel shaft emerges as shown in Fig. 1. The rim of a rubber drive roller on an arm is very lightly pressed against the motor shaft, and the tape rides over another section of the roller. For positive drive, the tape is pressed against the roller by a ball bearing at the end of another arm under spring pressure.

When the machine is still, the pressure of the rubber roller rim against the motor shaft is so light that no permanent indentation is made by the shaft. However, when the machinery starts, the fact that the roller is pulling the tape against some resistance at its other end (resulting from a felt pressure pad pressing the tape against the brass inertia wheel at the right) tends to snub the roller against the motor shaft, so that in use there is very positive engagement of rubber and motor shaft. This system, unique in tape mechanisms, to our knowledge, does away with any need for mechanical disengagement of the two when the Unit is shut off, and considerably simplifies the system.

The tape emerges from its cartridge at the right and passes up over the brass inertia wheel. This wheel runs free on its shaft and a felt pad on a spring arm presses the tape against it. It is this wheel's function to smooth out the variations in speed caused by a rotating system within the cartridge.

The tape next runs over the record head, three playback heads, and an erase head, in that order, at a speed of about 12 inches per second. The separation between heads is approximately uniform, and is such that a given point on the tape takes about 110 milliseconds (.110 second) to go from one head to the next. After passing over the rubber drive roller, the tape returns to the cartridge. Since endless tape is not available, the loop is made with a splice, using an unusual flexible splicing tape which does not appreciably add to the stiffness of the tape at the splice point.

A word of explanation about the selection of 110 milliseconds as the period between reverberatory repetitions, since this may seem a little long to some readers who have experience in the field. Many workers have felt that it is a prime necessity to keep the period between repetitions short enough so that under *no* circumstances can repetitions be heard separately rather than blending into an apparently continuous decay. This approach has some merit, and in a real auditorium there are so many different reverberations periods due to the many different trains of repetitions from various surfaces that discrete repetitions are almost never apparent.

It is an empirically proven fact, however, that even in the typical auditorium, which has so many complex reverberation paths, the ear seems to "select" the longest period between repetitions, which represents the reflecting surfaces which are farthest apart, and judges from that alone the size of the auditorium. And in an artificial system, if all the periods between repetitions are short the hearer gets an impression of a small room — which is unpleasing to begin with and may also sound highly unnatural if the system can at the same time give long decay times which would never occur in such a small room. This is one of the main failings of spring systems.

It is therefore far better to choose the longest period between repetitions which can be tolerated without causing discrete and separate sounds *in most cases,* because this sounds like a large room. A note must last at least 110 ms to avoid discrete repetitions in the Reverbatape system, but this is still a very short staccato, and the repetitions will be apparent only near the full-reverberation control settings. Faster music, which is likely to have larger numbers of staccato notes, would always call for a reduced reverberation-time setting to attain clarity. Then any discrete repetitions that may occur do not last long and are covered or masked by following music.

The compromise of 110 ms has proved over a long period and with the widest possible variety of music to be a good one.

It can, however, cause difficulty with some kinds of program material if improperly used. The sounds of speech are extremely staccato, and so are those of some instruments such as piano and harpsichord. When it is used for these sounds, the control must not be turned up too high. Oddly enough, when the Unit is used with a guitar amplifier, no problems arise. Evidently, the decay of sound after a guitar string is plucked is so smooth that it approximately follows the reverberative decay.

Electronic Requirements

The initial specifications upon which the total design is based were arrived at on a purely subjective basis. While the science of acoustics can tell us — at least in part — how to achieve certain reverberation characteristics in building a hall, the question of what these characteristics should be is

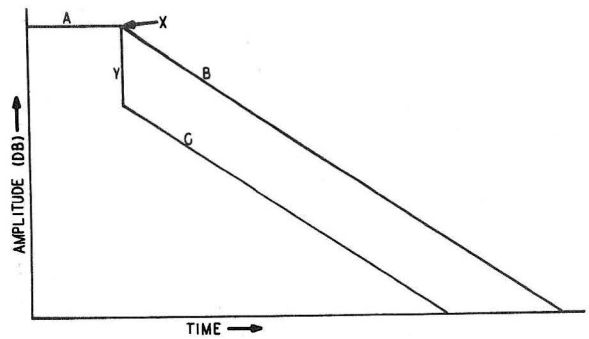

Fig. 2 — The unnatural effect of controlling a reverberator by simply reducing the relative gain of the reverberation channel. Abrupt drop in level occurs after sound source ceases, followed by decay with unchanged slope.

still a matter of what an accomplished and experienced musician feels is best. This subjective approach is entirely valid because music itself, being an art, is intended to affect the emotions. Man has not yet learned to analyze himself enough to make reliable correlations between scientific criteria, such as meter readings, and human emotional reactions.

The equipment used for determining the design objectives was a large, specially built machine mounted in a 6-foot rack. It had a large tape deck with means for driving tape at various

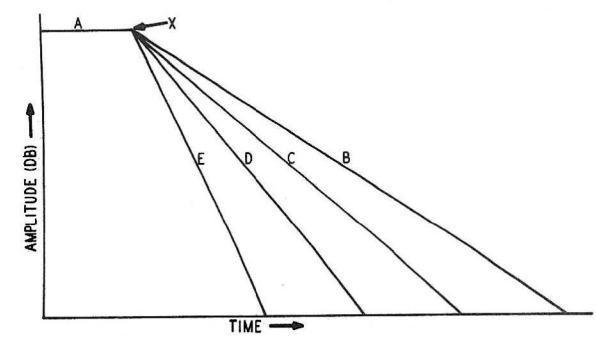

Fig. 3 — Natural decay characteristics for various reverberation times, as given by the REVERBATAPE Unit.

speeds and several record-playback heads which could physically be moved to obtain any desired separation between them. The rack had 6 amplifiers, a multitude of switches and controls for volume and equalization, and facilities for doing almost anything. In several months of experimenting with number of heads, head placements, tape speeds, equalization to determine relative reverberation times for various parts of the audio spectrum, and the like, the final requirements for the production units were obtained. Except to determine what had been achieved finally, meters were not used; for every test, organ and other music was actually played through the system and listened to by several sets of trained ears. Only two results were desired: the utmost realism; and the ability to control reverberation to fit various kinds of music while maintaining complete realism at each control setting.

It was discovered during this phase that the control methods usually used in artificial reverberation systems — especially the spring units — was unrealistic. Usually there is a direct channel, through which sound goes without added effect, and a parallel reverberation channel, in which reverberative repetitions are added to the original. These are combined at their outputs. To reduce reverberation time, a control simply reduces the gain of the reverberation channel, so that relatively more direct sound is heard.

Fig. 2 shows the result. With full reverberation setting, the direct sound is represented by line A. The sound source ceases (the player releases a key) at time X. Then the reverberation channel causes a slow decay along line B.

When the reverberation channel is turned down, the direct sound A stops at time X. There is then an abrupt drop in level (Y), after which the decay occurs at the same rate (C) as before. No real auditorium does this. Instead, the decay always begins smoothly just when the sound source is removed. Fig. 3 shows this effect at various total reverberation times. Each decay curve B, C, D, E, is a smooth one beginning exactly at point X, without any abrupt drop after cessation of source. The decay slopes are, of course, different. To accomplish this necessary bit of realism in the Reverbatape Unit a new kind of attenuation system had to be designed — one which would

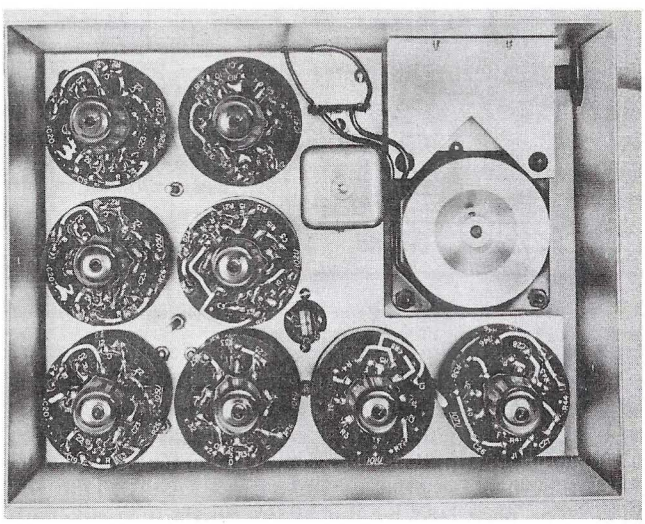

Fig. 4 — Looking at rear of main unit with phenolic cover removed shows the 8 tubes, each atop its plug-in module.

place output level at each reverberator repetition a certain number of decibels less than the last at all control settings, the certain number changing with control settings. This is described in detail later.

Being a tape recorder, the unit requires the usual record amplifier, bias-erase oscillator, and a playback amplifier for each playback head. This adds up to five dual-triode stages. In addition, there is an input stage, which also controls feedback from the last playback head into the record amplifier so the signal will repeat continually, a mixer stage for the head amplifier outputs, and a cathode-follower output stage. Each stage is mounted on a special plug-in module which can quickly be replaced should servicing be necessary. The rear photo of Fig. 4 shows the modules in place, with the phenolic rear cover removed.

Circuitry

Fig. 5 is a complete schematic diagram of the Reverbatape Unit, minus the power supply, which is a separate small unit. Each of the plug-in modules is outlined by a dashed line, with a number (16101, etc.) to identify it.

The Unit is intended to have unity gain, and in use it is connected between the organ and its regular amplifier, or between two existing stages in a voltage amplifier. Input impedance is 1 megohm. A special plug-in attenuator contains voltage dividers for both input and output, so that, whatever the signal level available from the organ, the record amplifier can be almost fully modulated for best signal-to-noise ratio.

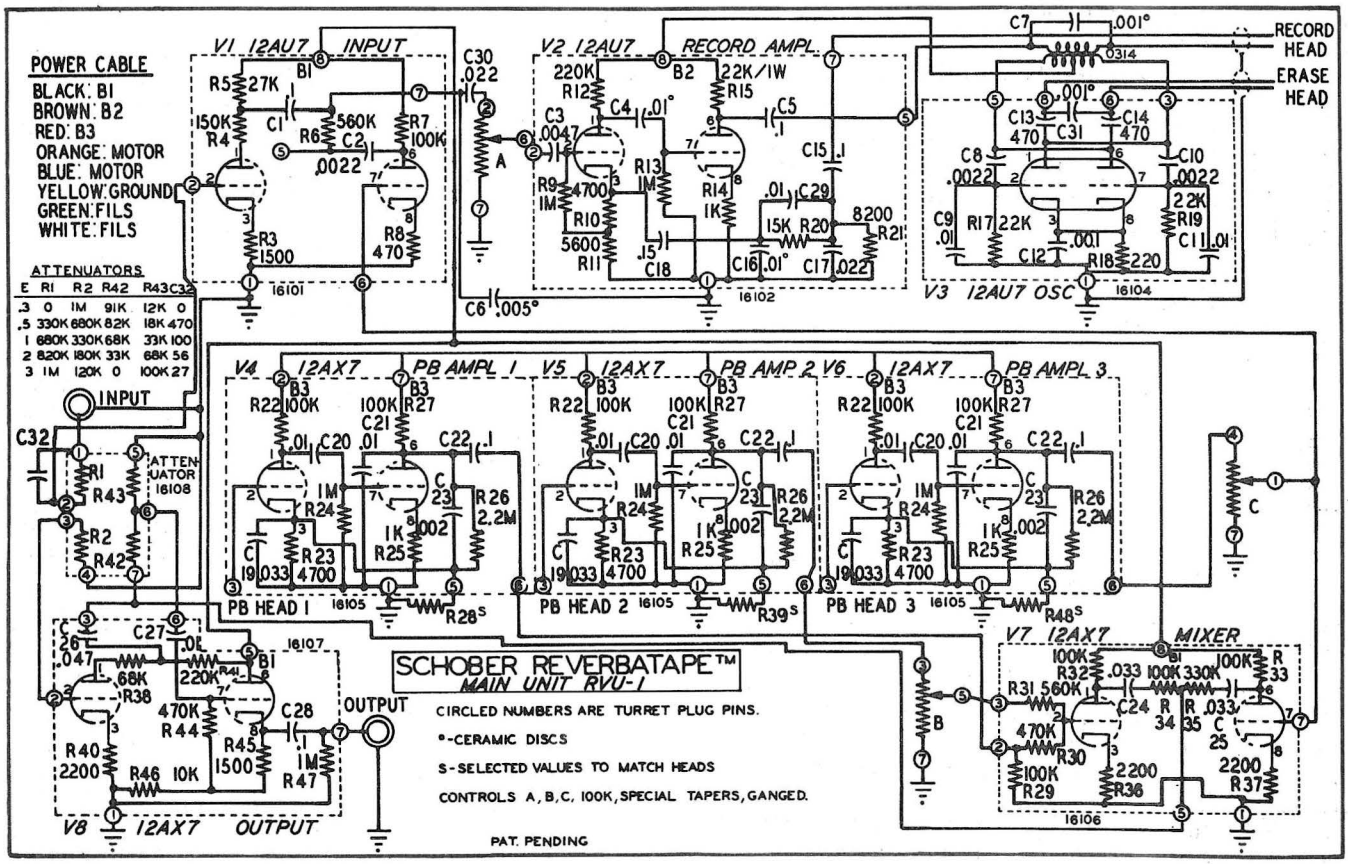

Fig. 5 — Complete schematic (except power supply). Each module is enclosed in dashed lines and numbered.

Input signal goes through the input attenuator R1-R2 (compensated by C32 for stray capacitive treble losses) to the grid of V8. The plate load resistor of this triode is in two parts R38-R41 (B1 indicates one of the B+ lines) and signal from the junction is applied to the output attenuator R42-R43, from the junction of which it is sent to the second grid of V8, the cathode follower which furnishes the system output. Both attenuators and the plate-resistor division are always so proportioned that the output voltage is about equal to the input. This is the direct channel and it does not alter the signal in any way.

Input signal from the same input attenuator is also sent to the first grid of V1. Here again there is a divided plate resistor; signal at the junction goes through potentiometer A to the grid of recording amplifier V2. (Disregard the second triode of V1 for the moment.) The record amplifier and the bias oscillator V3 are conventional, the former containing the usual treble-emphasis equalizer as part of a feedback path taken between the "cold" terminal of the record head and ground and returned to the first V2 cathode. Potentiometer A is part of the special 3-gang panel control, which will be explained later in detail.

V4, V5, and V6 are identical playback amplifiers, one for each playback head, each containing a feedback-type bass-boost equalizer. R28, R39, and R48 are calibration resistors which are selected at the factory so that the outputs of the three amplifiers are uniform despite the usual manufacturing tolerances of the heads.

V7 is a mixer stage in which the three playback signals are mixed in definite proportions and then fed to the output attenuator along with the direct signal. The combined direct and played-back signals go then to the grid of the cathode follower and out.

Control System

The control system is one of the most important Reverbatape features, and it is new. (A patent is pending.) The pipe organist in a church is stuck with whatever reverberation characteristics the church happens to have, but no one characteristic can suit all kinds of music. Reverberation time is defined as the time required for a tone of 512 cycles per second (an octave above middle C) to decay 60 decibels. If this takes perhaps 6 seconds, the hearer receives an aural impression of a very large and impressive auditorium. However, such a time is suitable only for very slow, stately music. Faster music becomes blurred and unclear because while any notes are being played the sound of the notes played almost 6 seconds ago are still present. An extreme and very well known case is the Cathedral of St. John the Divine in New York, where the reverberation time is around 9 seconds. The very fine pipe organ installed there is almost useless. Nonetheless, long reverberation times are sometimes desired.

The Reverbatape Unit is so calibrated that when the control is at maximum the output of the Unit due to signal from the first playback head is about 1 db less than the output due to the direct signal. Output due to the second head is

1 db less than that, and signal due to the third head is still another decibel lower. Output of the third head is also fed back to the record amplifier and rerecorded, so that this cycle continues — each succeeding repetition 1 db less than the last.

With this control setting, and with the repetitious spaced 0.11 second apart, it takes 60 repetitions or 6.6 seconds for an attenuation of 60 db, and this is the maximum reverberation time of the device. (This is, of course, theoretical; actual calibrations may vary somewhat.)

Faster music, or music requiring greater clarity, however, requires shorter reverberation times. These are obtained when the control is rotated counterclockwise by increasing the attenuation between played-back repetitions. For example, when the setting gives 2 db between repetitions, only 30 repetitions or 3.3 seconds, is needed to reduce level by 60 db. The control is continuous from maximum down to zero. The following explanation of how it works requires careful attention to avoid confusion.

Maximum reverberation time is obtained with the sliders of controls A, B, and C all the way up. Control A then sends full signal to the recording amplifier. Signal from head amplifiers 1 and 2 are joined through R30 and R31 and go to the first mixer grid. The resistors are so proportioned that the head-2 signal reaching the grid is 1 db less than that of head 1. The head-3 signal goes to the second mixer grid. The two plate outputs are mixed through R34 and R35,

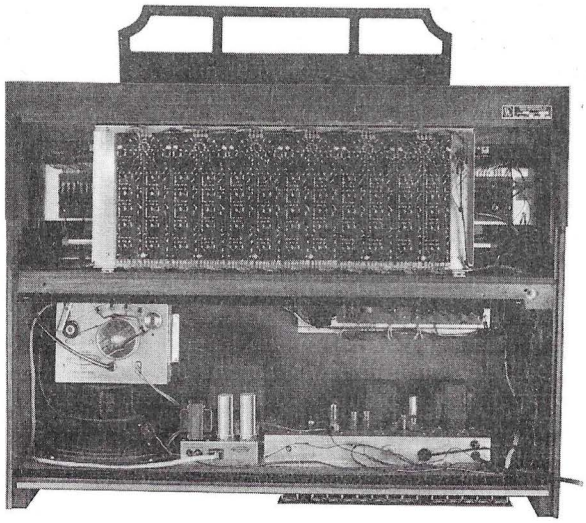

Fig. 6 — A typical installation in an organ. Main Unit is attached to wood with its side brackets; Power Supply is screwed down by flanges.

whose values determine that at the junction (which leads to the output stage) the head-3 signal is 1 db less than that of head 2. The three head outputs, therefore, are all 1 db apart, satisfying the conditions given above. This relative attenuation is permanent.

Let us specify certain terminology. Since the values in the output stage V8 determine that the first head signal appears 1 db lower than the direct signal, let us arbitrarily refer always to the output due to the direct channel as 0 db. With the control at maximum, playback signal 1 is then —1, 2 is —2, and 3 is —3.

Head-3 output from control C is fed to the second V1 triode, as well as to the mixer. The values here are so chosen that when this fed-back signal joins the input signal through R6 and arrives at the top of control A, it has an actual value 3 db less than the signal due to the system input. We may consider this 3 db of attenuation permanent. Since the direct signal, recorded and played back through head 1, produced an original output of —1 db, the fed-back signal produces a result 3 db less, or —4 db. This is 1 db less than the output due to the original head-3 signal, and it continues the scheme of continual 1-db-apart repetitions.

We may also consider that due to the mixer values, the output of head 2 has a permanent 1-db attenuation, and that of head 3 has a permanent 2-db attenuation. Adding 3 db to each of these for the second round of signals produced as the result of the feedback gives for head 2 3 db less than the original —2 (—5), and for head 3 3 db less than the original —3 (—6). Thus the action continues, repetitions at the output always 1 db apart.

So far, the three ganged control potentiometers have always been at maximum. Before moving them, we must explain that they have special tapers, so that at any position the

Fig. 7 — The REVERBATAPE Control may often be mounted on the organ stop board; a decal is supplied.

attenuation given by control C is always twice that (in decibels) of controls A or B, which are identical. Thus, if the control is set so that A attenuates its signal 3 db, then control B has 3 db of attenuation and control C attenuates by 6 db.

Let us now reduce the control setting so that A and B have 1 db of attenuation each — and C has 2 db.

The input signal passes through control A and suffers 1 db of attenuation. The output of head 1 is thus 1 db less than with the control wide open. Instead of appearing at the system output as —1, it appears at —2.

The output of head 2 also is 1 db less than before. It also suffers another 1 db of attenuation in control B, making a total of 2 db. Therefore, instead of being at —2 as originally, it is now at —4.

The output of head 3 is also 1 db less than before. It also is attenuated 2 db more by control C, making a total of 3. So instead of appearing at the output at —3, it appears at —6.

You will note, therefore, that at this control setting, the

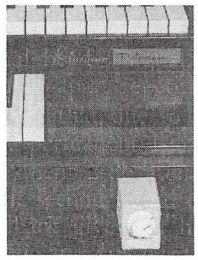

Fig. 8 — Where mounting the Control on a wood board is impossible or not preferred, the plastic control box is mounted below the lower manual with a special removable adhesive pad.

system outputs due to the three heads are respectively —2, —4, and —6 with respect to our standard 0-db direct channel. The reverberation time is reduced, but the decay curve, composed of equal decrements in repetitions, is still linear.

Now let us consider the fed-back signal with this control setting. Since the signal to the recording head is attenuated 1 db by control A and control C attenuates the head-3 signal by 2 db more, the fedback signal arrives at the second V1 grid 3 db lower than it was before. In going to the recording amplifier it passes through control A, for 1 more decibel of extra attenuation. It therefore comes to the record amplifier atten-

uated by a total of 4 db more than it did when the control was wide open. When the control was open, the fed-back signal produced a system output of —4 db due to head 1. Subtracting 4 db from that, the fed-back signal with the new control setting, due to head 1, is —8 db, which continues the cycle of 2-db decrements as advertised. For a little mental exercise, you can postulate other control settings and additional feedback cycles; you will find that decrements between repetitions are always equal!

Reverbatape Installation

To add the Reverbatape Unit to an audio system of any kind, whether it be an electronic organ or a high-fidelity, recording, or broadcast system, it is only necessary to find a point in the amplification section where the maximum audio voltage is something between about 0.3 and 3.0 volts rms. Such a point may be between two voltage-amplifier stages, for instance. Or in a Schober Organ or a high-fidelity system, it may be between the organ or preamplifier output and the power amplifier. The circuit is broken at this point (usually by using existing plugs and sockets or a tube-socket adapter). For example, in a Schober Organ, the organ output, which is normally fed directly to a power amplifier (or to the "auxiliary" position of a high-fidelity system function switch), is fed instead to the input jack of the Reverbatape Unit. Then the output jack of the Unit is fed to the amplifier. A survey of the organ field has made it possible to prepare instruction sheets covering each make and model of organ, showing physical placement of the parts, and the connections. In most cases, the necessary cords and connectors are furnished already made up in a small separate Installation Kit tailored for the particular organ.

The Unit box is attached to any convenient surface within the organ or equipment cabinet with a pair of universal brackets, and the power supply is fastened near it. Fig. 6 shows a typical Reverbatape installation.

The control may be fastened directly to the organ stop board, as in Fig. 7, or to any wood or metal panel. A plastic control box is also available, for easier installations. This can be fastened to the organ just below the lower manual (Fig. 8), usually without screws and without marring the woodwork.

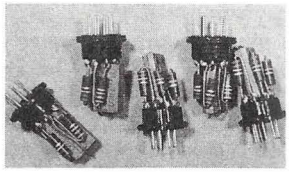

Fig. 9 — These are the attenuators. They improve signal/noise ratio by adapting the Unit to the audio level available from the organ or audio system at the point of insertion.

In any installation, maximum signal-to-noise ratio is obtained by using the correct plug-in attenuator (module 16108 in Fig. 5). If specific instructions are not available for the particular organ or equipment, the smallest-voltage attenuator which can be used without distortion on loud passages is selected, either by trial or by voltage measurement. This allows the maximum audio present to modulate the recording system fully. A few of these little attenuators are shown in Fig. 9.